IN THE DRAWINGS

Please amend FIG. 3 to be replaced with the enclosed replacement sheet. It is believed that no new matter is included in the submitted replacement sheet as it only clarifies a reference number. The enclosed drawing sheet is labeled "Replacement Sheet" as required by current drawing amendment procedures.

REMARKS

In the non-final 3 April 2007 *Office Action*, the Examiner rejects Claims 1-5, 7-9, 11-27, and 29-54. Applicants thank the Examiner with appreciation for the careful consideration and examination given to the Application.

Applicants file this response solely to facilitate prosecution. As such, Applicants reserve the right to present new or additional claims in this Application that have similar or broader scope as originally filed. Applicants also reserve the right to present additional claims in a later-filed continuation application that have similar or broader scope as originally filed.

After entry of this Response, 1-5, 7-8, 11-23, 25-27, and 29-54 are pending in the Application. Applicants respectfully assert that the pending claims are in condition for allowance and respectfully requests reconsideration of the claims in light of the following remarks.

I. Drawing & Specification Amendments

Applicants amend the reference numeral for "compensation filter" to 43, as numeral 33 was used for both "synchronizer" and "compensation filter," in the above-presented *Specification* paragraphs. Also, Applicants also amend Figure 3 to include reference number 43.

II. Claim Amendments

Applicants amend Claims 1-5, 7-8, 11, 14-23, 26-27, 29-47, and 49-54 to better describe the present invention. Claims 9 and 24 are cancelled without prejudice or disclaimer. Exemplary support for amended Claim 1 may be found in at least the following places in the originally filed *Specification*:

"generating a first predictable noise" at page 13, line 22, and Figure 3, numeral 31a; "converting the first predictable noise to an audio output using a converter having a known transfer function" at page 13, lines 23-25, and Figure 3, numeral 31b; "receiving the audio output at the first microphone" at page 13, line 28, and Figure 3, numeral 21a; "converting the audio output to a first output noise" at page 12, line 21, and Figure 3, numeral 23a; "generating a second predictable noise" at page 14, lines 8-9, and Figure 3, numeral 32; "synchronizing the first predictable noise and the second predictable noise in time by a synchronizer" at page 14, lines 9-12, and Figure 3, numeral 33; "compensating the second predictable noise for the known

transfer function by a compensation filter“ at page 14, lines 12-14, and Figure 3, numeral 43; “outputting a second output noise by the compensating filter“ at page 14, line 8, and Figure 3, numeral 43; “determining coefficients representing a transfer function of the first microphone based on the first and second output noises“ at page 14, line 20 to page 15, line 7, and Figure 3, numeral 35; “determining a filtering function for the first microphone, wherein the product of the transfer function and the filtering function is a single selected function, and wherein the single selected function equals a second product of a second transfer function and a second filter function of the second microphone“ at page 15, lines 8-12, Figure 3, numeral 36; and page 10, line 27 to page 11, line 14; and “outputting the coefficients to an equalization filter“ at page 15, line 12, and Figure 3, numeral 37.

Independent claims 16, 42, and 43 are directed to an apparatus, a sound system, and a method for using a sound system, and generally parallel independent claim 1. Thus, it is believed the above-recited *Specification* passages also fully support amendments to Claims 16, 42, and 43.

Accordingly, the presently introduced clarifying claim amendments are fully supported by the application as originally filed. Thus, it is believed that no new matter has been introduced by way of the amendments to the claims.

III. All Pending Claims Are Patentable In Accordance With 35 U.S.C. § 112

The Office rejects Claims 1, 16, 42, 43, and 51-54 under 35 U.S.C. §112, first paragraph, for allegedly failing to comply with the enablement requirement. Specifically, the Office asserts that these claims contain subject matter not described in the specification.

Applicants respectfully disagree. To expedite the prosecution of the present application, however, Applicants amends Claims 1, 16, 42, and 43. Exemplary support for the claims is provided above. As such, Applicants respectfully request the Office to withdraw the §112 rejection.

IV. The Objections To Claims 39-41 Are Overcome

Claims 39-41 have been amended to be dependent on one single claim. Thus, Applicants respectfully request the Office to withdraw the objection to Claims 39-41.

V. All Pending Claims Are Patentable In Accordance With 35 U.S.C. § 103

The Office rejects Claims 1-4, 7-9, 14-19, 22-24, 26, 29-31, 33-45, and 51-54 under U.S.C. §103(a) as allegedly being unpatentable over Vaughn (US Patent No. 5,233,655), hereinafter referred to as Vaughn, in view of Hamabe (US Patent No. 5,426,703), hereinafter referred to as Hamabe.

The Office further rejects Claims 5, 12-13, 20-21, 25, 27, 32, and 46-48 under U.S.C. §103(a) as allegedly being unpatentable over Vaughn in view of Hamabe, further in view of T. Schneider et al., J. Audio Eng. Soci., Vol. 41, No. 7/8 1993, pp. 583-593, hereinafter referred to as Schneider.

The Office further rejects Claims 49-50 as being unpatentable over Vaughn in view of Hamabe, further in view of R. A. Roberts et al., "Digital Signal Processing" pp. 486-489.

Claims 9 and 24 have been cancelled, thus rendering the Office's rejection to these claims moot.

Applicant respectfully requests reconsideration and withdrawal of the rejections in view of the amendments made herein. The applied references fail to disclose or suggest the inventions defined by Applicants' claims, and provide no teaching that would have suggested the desirability of modification to arrive at the claimed inventions.

The present application discloses a method, an apparatus and sound system of equalizing output signals from a first microphone and a second microphone by generating a first predictable noise. The first predictable noise is converted to an audio output using a converter having a known transfer function. The audio output is received at the first microphone; and converted to a first output noise. A second predictable noise is generated which is synchronized to the first predictable noise in time, by using a synchronizer. The second predictable noise is compensated for the known transfer function by a compensation filter. Coefficients representing a transfer function of the first microphone are determined based on the first and second output noises. A filtering function is determined for the first microphone, so that the product of the transfer function (M) and the filtering function (H) is a single selected function (F), which equals a second product of a second transfer function and a second filter function of the second microphone, i.e. M1*H1=F and M2*H2=F. See page 11, lines 2-23 of the description.

Vaughn teaches an audio equalizer "network" with a set of 24 analog transversal bandpass filters having a passband extending continuously across the audio spectrum. The group

delay of the filter set varies continuously with frequency across the audio spectrum, with the group delay being relatively high at frequencies less than approximately 200 Hz and being relatively low but greater than or equal to zero at frequencies greater than approximately 8000 Hz.

Compared to the claimed invention, Vaughn at least does not teach or suggest the following claimed features:

<u>A. "synchronizing the first predictable noise and the second predictable noise in time by a synchronizer"</u>

Vaughn does not teach or suggest the generation of a second predictable noise which is synchronized in time to the first predictable noise. Vaughn teaches a white noise 44a and a pink noise 44b which may be selected by a signal selector/combiner 26 to be input into the bandpass filters. See Figure 1, column 5, line 59, and column 6, lines 8-10 of Vaughn. It should be apparent to a person skilled in the art that a pink noise is a signal with a frequency spectrum such that the power spectral density is proportional to the reciprocal of the frequency, and cannot be synchronized to a white noise in time. The Office suggests at page 11, line 4 of the Office Action that the microprosessor controller 45 of Vaughn is a synchronizer, but this interpretation is unreasonable and has no support. Vaughn does not teach or suggest the synchronization between any two signals, nor any synchronizer. In fact, Vaughn never used any word with a word stem "synchron-", including "synchronizer."

<u>B. "compensating the second predictable noise for the known transfer function by a compensation filter"</u>

Vaughn does not teach or suggest a compensation filter compensating the second predictable noise for the known transfer function of the first predictable noise as in the present claimed invention. As discussed above, Vaughn does not teach or suggest a second predictable noise which is synchronized to the first predictable noise. Furthermore, the numeral 26 in Figure 1 of Vaughn is a programmable signal selector/combiner, which is used by a therapist to apply signals. See Figure 1, column 5, lines 59-60, and column 9, lines 53-59 of Vaughn.

C. “determining coefficients representing a transfer function of the first microphone based on the first and second output noises” and “outputting the coefficients to an equalization filter”

As Vaughn does not teach or suggest digital signal processing (DSP) technology for processing the audio signals, it does not consider a microphone in terms of a transfer function. Accordingly, Vaughn does not teach or suggest coefficients representing a transfer function.

The Office appears to interpret numeral 10 in Figure 1 of Vaughn as a transfer function at page 11, second paragraph of the Office Action. This is not correct. Numeral 10 in Figure 1 of Vaughn is the whole hearing aid system. See Figure 1, and column 5, line 51 of Vaughn:

D. “determining a filtering function for the first microphone, wherein the product of the transfer function and the filtering function is a single selected function, and wherein the single selected function equals a second product of a second transfer function and a second filter function of the second microphone”

Applicants submit that Vaughn does not teach or suggest a filtering function such that the product of the transfer function and the filtering function is a single selected function.

E. The Secondary References Do Not Cure Hamabe’s Many Deficiencies

Hamabe offers nothing to overcome the basic deficiencies of Vaughn and the other references of record. Furthermore, as submitted in the previous response, in Hamabe the audio output from the loudspeaker varies with the time due to the properties of the environmental noise, hence identification of the microphone transfer function is not possible by Hamabe. As such, Applicants respectfully submit that the currently pending claims are patentable over the cited combinations.

Applicants respectfully submit that the dependent claims are inventive at least by virtue of their dependencies and further distinguish the invention. The rejections to the claims are now moot and do not, therefore, need to be addressed individually at this time. It will be appreciated, however, that this should not be construed as Applicants acquiescing to any of the purported teachings or assertions made regarding the cited references.

VI. Fees & Express 37 CFR § 1.136 Petition For Extension

This *Response and Amendment* is being filed within five months of the 3 April 2007 *Office Action*. Thus Applicant petitions for a two-month extension pursuant to 37 CFR § 1.136 and the undersigned submits the applicable fee via EFS-Web. No claims fees are believed due because no new claims are presented. Applicants believe that no additional fees are due, however, the Commissioner is authorized to charge any other fees or credit any overpayments to Deposit Account No. 20-1507.

VII. Conclusion

The foregoing is believed to be a complete response to the *Office Action* mailed 3 April 2007. Applicant respectfully asserts that the pending claims are in condition for allowance and respectfully requests passing of this case in due course of patent office business. If the Examiner believes there are other issues that can be resolved by a telephone interview, or there are any informalities remaining in the application which may be corrected by an Examiner's amendment, a telephone call to Hunter Yancey at (404) 885-3696 is respectfully requested.

Respectfully submitted,

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APPENDIX A: Clean Listing of Pending Claims

1. A method of equalizing output signals from a first and a second microphones, the method comprising the steps of:

generating a first predictable noise;

converting the first predictable noise to an audio output using a converter having a known transfer function;

receiving the audio output at the first microphone;

converting the audio output to a first output noise;

generating a second predictable noise;

synchronizing the first predictable noise and the second predictable noise in time by a synchronizer;

compensating the second predictable noise for the known transfer function by a compensation filter;

outputting a second output noise by the compensating filter;

determining coefficients representing a transfer function of the first microphone based on the first and second output noises;

determining a filtering function for the first microphone, wherein the product of the transfer function and the filtering function is a single selected function, and wherein the single selected function equals a second product of a second transfer function and a second filter function of the second microphone; and

outputting the coefficients to an equalization filter.

2. A method according to claim 1, wherein the single selected function is the transfer function of one of the first and second microphones.

3. A method according to claim 1, wherein the single selected function is a common factor.

4. A method according to claim 1, further comprising the step of:

loading the filtering function to the equalization filter.

5. A method according to claim 1, wherein the first predictable noise is a first predictable noise sample signal

the second predictable noise is a second predictable noise sample signal and wherein the second predictable noise sample signal has a property substantially identical to the first predictable noise sample signal.

6. (Cancelled)

7. A method according to claim 1 further comprising the steps of:

providing a propagation time delay for the first predictable noise before the first microphone converting the first predictable noise sample to the first output noise;[[,]] and

delaying the second output noise by same amount of time as the propagation delay time.

8. A method according to claim 7, wherein the first predictable noise signal is a first predictable digital noise signal, and the second predictable noise signal is a second predictable digital noise signal.

9 -10. (Cancelled)

11. A method according to claim 7, wherein the propagation delay time is an integer multiple of the first predictable noise sample.

12. A method according to claim 8, wherein the step of generating the first predictable digital noise signal includes a step of utilizing a maximum length sequence generator to generate the first predictable digital noise signal.

13. A method according to claim 8, wherein the step of generating the second predictable digital noise signal includes a step of utilizing a maximum length sequence generator to generate the second predictable digital noise signal that is substantially identical to the first predictable digital noise signal on a sample-by-sample basis.

14. A method according to claim 8, wherein the first predictable digital noise signal or the second predictable digital noise signal comprises a white noise signal.

15. A method according to claim 8, wherein the first predictable digital noise signal or the second predictable digital noise signal comprises a random noise signal.

16. An apparatus for equalizing output signals from a first and a second microphones, the apparatus comprising:

a first generator generating a first predictable noise;

a first converter converting the first predictable noise to an audio output, the first converter having a known transfer function, the first microphone receiving the audio output;

a second converter converting the audio output to a first output noise;

a second generator generating a second predictable noise;

a synchronizer synchronizing the first generator and the second generator,

a compensation filter compensating the known transfer function of the first converter,, the compensation filter outputting a second output noise based on the compensation;

an identification circuit determining coefficients representing a transfer function of the first microphone based on the first and second output noises;

a determination circuit determining a filtering function for the first microphone, wherein the product of the transfer function of the microphone and the filtering function is a single selected function, and wherein the single selected function equals a second product of a second transfer function and a second filter function of the second microphone; and

an equalization filter receiving the coefficients.

17. An apparatus according to claim 16, wherein the single selected function is the transfer function of one of the first and second microphones.

18. An apparatus according to claim 16, wherein the single selected function is a common factor.

19. An apparatus according to claim 16, further comprising: a loader loading the filtering function to the equalization filter.

20. An apparatus according to claim 16, wherein the first predictable noise is a first predictable noise sample signal; and wherein the second predictable noise is a second predictable noise sample signal, and wherein the second predictable noise sample signal has a property substantially identical to the first predictable noise sample signal.

21. An apparatus according to claim 20, further comprising an analog-to-digital converter coupled to the microphone converting an electrical analog signal of the first microphone into a digital signal.

22. An apparatus according to claim 16, further comprising:
a module providing the first predictable noise with a propagation time delay, before the first microphone converting the first predictable noise sample to the first output noise[[,]]; and
a second module providing a second predictable noise with the propagation time delay.

23. An apparatus according to claim 22, wherein the first predictable noise signal is a first predictable digital noise signal, and the second predictable noise signal is a second predictable digital noise signal, further comprising:

a noise generator for generating the first predictable digital noise signal and the second predictable digital noise signal.

24. (Cancelled)

25. An apparatus according to claim 23, wherein the noise generator includes a maximum length sequence generator for generating the first predictable digital noise signal that is substantially identical to the second predictable digital noise signal on a sample-by-sample basis.

26. An apparatus according to claim 16, wherein the first converter includes a loud speaker.

27. An apparatus according to claim 23, wherein the first predictable digital noise signal is a first maximum length sequence noise, and wherein the second predictable digital noise signal is a second maximum length sequence noise being substantially identical to the first maximum length sequence noise on a sample-by-sample basis.

28. (Cancelled)

29. An apparatus according to claim 22, wherein the propagation delay time is an integer multiple of the first predictable noise sample.

30. An apparatus according to claim 23, wherein the first predictable digital noise signal or the second predictable digital noise signal comprises a white noise signal.

31. An apparatus according to claim 23, wherein the first predictable digital noise signal or the second predictable digital noise signal comprises a random noise signal.

32. An apparatus according to claim 23, wherein the noise generator includes a maximum length sequence generator for generating the first predictable digital noise signal and the second predictable digital noise signal.

33. A method for equalizing two or more microphones in a listening devices using the method according to claim 1.

34. A method for equalizing two or more microphones in a hearing aid according to claim 1.

35. A method for equalizing two or more microphones in a headset according to claim 1.

36. An apparatus according to claim 16, wherein the apparatus is a listening device.

37. An apparatus according to claim 16, wherein the apparatus is a hearing aid.

38. An apparatus according to claim 16, wherein the apparatus is a headset.

39. A listening device according to claim 36, further comprising: a signal equalization filter provided for each of the first and the second microphones, wherein the function of the signal equalization filter is determined by the apparatus according to claim 36 and is loaded to the signal equalization filter.

40. A hearing aid according to claim 37, comprising: a signal equalization filter provided for each of one or more microphones, wherein the function of the signal equalization filter is determined by the apparatus according to claim 37 and is loaded to the signal equalization filter.

41. A headset according to claim 38, further comprising: a signal equalization filter provided for each of the first and the second microphones, wherein the function of the signal equalization filter is determined by the apparatus according to claim 38 and is loaded to the signal equalization filter.

42. A method of providing sound signals to a user through a system including two or more microphones, the method comprising steps of:

preparing a filtering function for each of one or more microphones, including, for each of the two or more microphones, the steps of:

generating a first predictable noise;

converting the first predictable noise to an audio output using a converter having a known transfer function receiving the audio output at a first microphone

converting the audio output to a first output noise;

generating a second predictable noise;

synchronizing the first predictable noise and the second predictable noise in time by a synchronizer;

compensating the second predictable noise for the known transfer function by a compensation filter;

outputting a second output noise by the compensating filter;

determining coefficients representing a transfer function of the first microphone based on

the first and second output noises; and

determining a filtering function for the first microphone, wherein the product of the transfer function of the microphone and the filtering function is a single selected function, wherein the single selected function equals a second product of a second transfer function and a second filter function of the other members of the two or more microphones; andoutputting the coefficients to an equalization filter; and

operating the system, including the step of: for each of the two or more microphones, transferring a sound signal through the microphone and the equalization filter for the microphone.

43. A sound system for two or more microphones for transmitting sound signals, comprising:

a first generator generating a first predictable noise;

a first converter converting the first predictable noise to an audio output, the first converter having a known transfer function, wherein a first microphone of the two or more microphones receiving the audio output;

a second converter converting the audio output to a first output noise;

a second generator generating a second predictable noise;

a synchronizer synchronizing the first generator and the second generator,

a compensation filter compensating the known transfer function of the first converter, the compensation filter outputting a second output noise based on the compensation;

an identification circuit determining coefficients representing a transfer function of the first microphone based on the first and second output noises;

a determination circuit determining a filtering function for the first microphone wherein the product of the transfer function of the microphone and the filtering function is a single selected function,

and wherein the single selected function equals a second product of a second transfer function and a second filter function of other members of the two or more microphones; and an equalization filter receiving the coefficients.

44. A sound system according to claim 43, wherein the single selected function is the transfer function of one of the two or more microphones.

45. A sound system according to claim 43, wherein the single selected function is a common factor.

46. A sound system according to claim 43, wherein the first predictable noise is a first predictable noise signal; wherein the second predictable noise is a second predictable noise signal; and wherein the second predictable noise signal has a property substantially identical to the first predictable noise signal.

47. A sound system according to claim 46, wherein the first generator includes a maximum length sequence generator for generating the first predictable noise signal.

48. A sound system according to claim 47, wherein the maximum length sequence generator generates the second predictable noise signal.

49. An apparatus according to claim 16, wherein the identification circuit performs an Auto Regressive Moving Average (ARMA) to estimate the transfer function.

50. A sound system according to claim 43, wherein the identification circuit performs an Auto Regressive Moving Average (ARMA) to estimate the transfer function.

51. A method according to claim 1, wherein an output signal through the first microphone and the equalization filter for the first microphone is substantially equal to an output signal through the second microphone and the equalization filter for the second microphone with respect to phase or phase and magnitude.

52. An apparatus according to claim 16, wherein an output signal through the first microphone and the equalization filter for the first microphone is substantially equal to an output signal through the second microphone and the equalization filter for the second microphone with respect to phase or phase and magnitude.

53. A method according to claim 42, wherein the two or more microphones comprises at least a first microphone and a second microphone, and wherein an output signal through the first microphone and the equalization filter for the first microphone is substantially equal to an output signal through the second microphone and the equalization filter for the second microphone with respect to phase or phase and magnitude.

54. A system according to claim 43, wherein the two or more microphones comprises at least a first microphone and a second microphone, and wherein an output signal through the first microphone and the equalization filter for the first microphone is substantially equal to an output signal through the second microphone and the equalization filter for the second microphone with respect to phase or phase and magnitude.